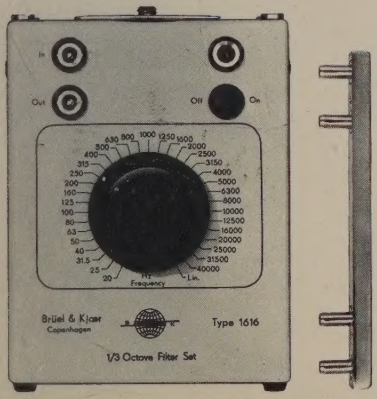
1616

Instructions and Applications



1/3 Octave Filter Set Type 1616

A portable third octave filter set which satisfies IEC Recommendation 225, ANSI S1. 11 – 1966 class III and DIN 45.652. It is designed for operation with and attachment to the B & K Sound Level Meters Types 2203 and 2209. Filter centre frequencies 20 Hz to 40 kHz, plus a „Linear" mode 22,4 Hz to 40 kHz.

BRÜEL & KJÆR

030-0467

1/3 OCTAVE FILTER SET
TYPE 1616

December 1973

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1. INTRODUCTION

The 1/3 Octave Filter Set Type 1616 is a compact, portable unit for 1/3 octave analysis in the range 18 Hz — 44 kHz covered by 34 pass-band filters. It is primarily designed for use in conjunction with the Precision Sound Level Meter Type 2203 or the Impulse Precision Sound Level Meter Type 2209. One screw is used to join the 1616 and the Sound Level Meter together, making a portable noise and vibration analyzer.

Filter centre frequencies are 20 Hz to 40 kHz arranged according to the preferred frequencies of ISO Recommendation R 266. Each filter satisfies the requirements of ANSI S1.11-1966 for third octave band filters of class III, IEC Recommendation 225, and DIN 45652.

A "linear" mode is also provided, giving a high pass characteristic with a very sharp low frequency cut-off at 22.4 Hz.

The Filter Set is battery powered from 3 standard type D cells (Type R 20 in IEC Publication 86-2).

2. CONTROLS

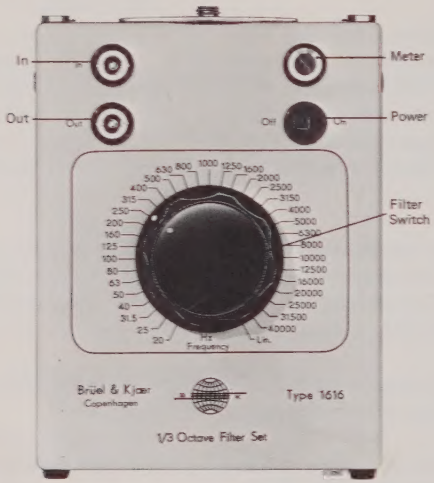

Fig.2.1. Front view of 1616

IN and OUT:

These sockets are for connection to the Sound Level Meter EXTERNAL FILTER sockets via the Connection Bar JP 0400 or to other equipment via cable AO 0007. The sockets accept the B & K coaxial plug JP 0006.

Input impedance at the IN socket is 10 kΩ, the maximum input voltage being 2 V peak. Max. DC voltage at input: 10 V. Output impedance at the OUT socket is less than 5 Ω in series with 0.47 μF. The socket

should not be loaded by less than 2 kΩ.

FILTER SWITCH:

A 35 position switch for selection of the required filter. Filter centre frequencies are indicated around the switch. A "Lin" position (22.4 Hz to 40 kHz) is also provided.

Before selection of a filter, the switch may be pushed in before turning. This introduces an attenuation of approx. 30 dB to the output signal so that transients due to switching do not overload the measuring device.

POWER:

On/Off switch for powering the instrument.

METER:

A built-in meter for battery check. Green and red areas indicate sufficient and insufficient battery condition. For changing batteries, see § 3.1.

3. OPERATION

3.1. REPLACEMENT OF BATTERIES

Before using the 1616, the battery condition should be checked. Set the POWER switch to "On". The needle of the built-in meter should deflect into the green area. If not, the batteries should be replaced. The batteries are accessible by sliding off the cover at the rear of the instrument (Fig.3.1). Take care to observe the correct polarity when inserting the new batteries.

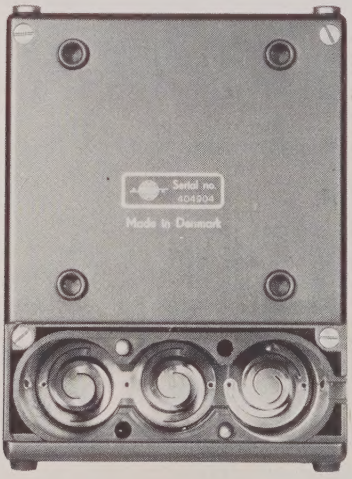

Fig.3.1. Rear view of 1616 with battery cover removed

3.2. MOUNTING ON THE SOUND LEVEL METER

The 1616 can be directly attached to either Precision Sound Level Meter Type 2203 or Impulse Precision Sound Level Meter Type 2209. For a 2203 with serial No. 424272 and earlier, a new bottom plate, part No. DD 0135, must be ordered separately to replace the original bottom plate.

The 1616 screws directly onto the Sound Level Meter using the large finger screw. Electrical connection is then established by plugging in the Connection Bar JP 0400 supplied with 1616. The portable analyzers thus obtained are shown in Fig.3.2 and 3.3.

If desired, move the leather carrying handle of the Sound Level Meter to give a correct balance of the combined instruments. Note that bushes for the handle's screws are on both sides of the instrument.

Fig.3.2. The 1616 mounted on the Precision Sound Level Meter Type 2203

Fig. 3.3. The 1616 mounted on the Impulse Precision Sound Level Meter Type 2209

Note: when the 1616 is fitted to the 2203, the Sound Level Meter batteries can only be changed by removing the Filter Set.

3.3. USE WITH THE PRECISION SOUND LEVEL METER TYPE 2203

3.3.1. Sound measurements

1. Calibrate the Sound Level Meter as described in its instruction manual.

2. Set the POWER switch of 1616 to "On" and check battery condition. The needle of the built-in meter should deflect into the green area. If not, change batteries as indicated in § 3.1.

3. Set the METER SWITCH of 2203 to one of the "Lin" positions ("Fast" or "Slow" as required).

4. Hold the instrument steadily as far away from the body as possible (preferably use a tripod or other firm, non-interfering support. The microphone should be at least 1 metre away from the observer's body) with the microphone directed towards the measuring object.

5. With the transparent OUTPUT ATTENUATOR knob in its fully clockwise position, turn the black INPUT ATTENUATOR knob until a meter deflection in the top 2/3 of the scale is obtained. If this is not possible, the transparent knob may also be used. However, the transparent knob should be kept as far clockwise as possible in order to maintain the optimum signal to noise ratio.

6. Set the METER SWITCH of 2203 to one of the "Ext. Filt." position ("Fast" or "Slow" as required).

7. Select the desired filter on the 1616 by depressing the FILTER SWITCH and turning it to the required position.

8. If the meter indicates less than 0 dB, increase sensitivity using the transparent OUTPUT ATTENUATOR on 2203.

9. The measured sound pressure level in the third octave band selected is the sum of the meter reading and the number in the red circle on the transparent knob.

3.3.2. Vibration measurements

1. Adjust 2203 for a K-factor of 0 as outlined in its instruction manual.

2. Set the POWER SWITCH of 1616 to "On" and check battery condition. The needle of the built-in meter should deflect into the green area. If not, change batteries as indicated in § 3.1.

3. Mount the accelerometer using one of the methods described in its instruction manual, taking care to avoid cable whip. For measurements of velocity or displacement, use can be made of the integrator ZR 0020 (see Fig.3.4).

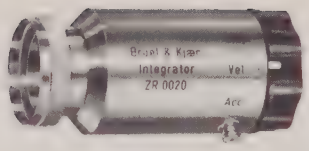

Fig.3.4. Integrator ZR 0020

4. Set the METER SWITCH of 2203 to one of the "Lin" positions ("Fast" or "Slow" as required).

5. Hold the instrument as far as possible from the vibration environment and other undesirable influences.

6. With the transparent OUTPUT ATTENUATOR knob in its fully clockwise position, turn the black INPUT ATTENUATOR knob until a meter deflection in the top 2/3 of the scale is obtained. If this is not possible, the transparent knob may also be used. However, the transparent knob should be kept as far clockwise as possible in order to maintain the optimum signal to noise ratio.

7. Set the METER SWITCH of 2203 to one of the "Ext. Filt." positions ("Fast" or "Slow" as required).

8. Select the desired filter on the 1616 by depressing the FILTER SWITCH and turning it to the required position.

9. If the meter indicates less than 0 dB, increase sensitivity using the transparent OUTPUT ATTENUATOR on 2203.

10. Read the vibration level in dB indicated by the meter.

11. Set the Slide Rule QH 0001 supplied with the Integrator to the correct accelerometer sensitivity (mV/g). An example is given in Fig.3.5.

12. Set the cursor to the number of dB read on the 2203 and read off the corresponding RMS acceleration, velocity or displacement.

Note that the slide rule cannot be used for conversion of for example acceleration to velocity. Each scale must only be used in connection with the appropriate setting of "Acc.", "Vel." or "Disp." on the Integrator ZR 0020.

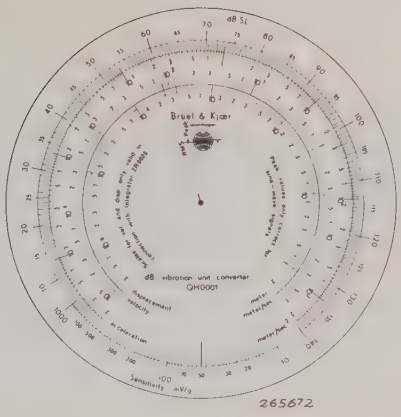

Fig.3.5. The Slide Rule set to an accelerometer sensitivity of 50 mV/g and an instrument reading of 60 dB

3.4. USE WITH THE IMPULSE PRECISION SOUND LEVEL METER TYPE 2209

3.4.1. Sound measurements

1. Calibrate the 2209 as described in its instruction manual.

2. Set the POWER SWITCH of 1616 to "On" and check battery condition. The needle of the built-in meter should deflect into ghe green area. If not, change batteries as indicated in § 3.1.

3. Set the controls of 2209 to:

WEIGHTING NETWORK	"Lin"
METER SWITCH	as required

4. Hold the instrument steadily, as far away from the body as possible (preferably use a tripod or other firm non-interfering support. The microphone should be at least 1 metre away from the observer's body) with the microphone directed towards the measuring object.

5. With the transparent OUTPUT ATTENUATOR knob in its fully clockwise position, turn the black INPUT ATTENUATOR knob until a meter deflection in the top 2/3 of the scale is obtained. If this is not

possible, the transparent knob may also be used. However, the transparent knob should be kept as far clockwise as possible in order to maintain the optimum signal to noise ratio.

If the OVERLOAD INPUT lamp flashes, the black INPUT ATTENUATOR knob should be switched anti-clockwise until the overload is removed. The meter deflection should then be restored by turning the transparent knob anti-clockwise.

6. Set the WEIGHTING NETWORK switch of 2209 to "Ext. Filter".

7. Select the desired filter on 1616 by depressing the FILTER SWITCH and turning it to the required position.

8. If the meter indicates less than 0 dB, increase sensitivity using the transparent knob.

   If the OVERLOAD OUTPUT lamp flashes, turn the transparent OUTPUT ATTENUATOR clockwise until the overload is removed. If the overload cannot be removed while retaining a meter deflection, any reading taken will be lower than the true signal level and the overload indication should be noted with the results.

9. The measured sound pressure level in the third octave band selected is the sum of the meter reading and the number which appears between the two red lines on the transparent knob.

3.4.2. Vibration measurements

1. Calibrate the 2209 as described in its instruction manual.

2. Set the POWER SWITCH of 1616 to "On" and check battery condition. The needle of the built-in meter should deflect into the green area. If not, change batteries as indicated in § 3.1.

3. Mount the accelerometer using one of the methods described in its instruction manual, taking care to avoid cable whip. For measurements of velocity or displacement, use can be made of the Integrator ZR 0020 (see Fig.3.4).

4. Set the controls of 2209 to:

WEIGHTING NETWORK	'Lin"
METER SWITCH	as required

5. Hold the instrument as far as possible from the vibration environment and other undesirable influences.

6. With the transparent OUTPUT ATTENUATOR knob in its fully clockwise position, turn the black INPUT ATTENUATOR until a meter deflection in the top 2/3 of the scale is obtained. If this is not possible, the transparent knob may also be used. However, the transparent knob should be kept as far clockwise as possible in order to maintain the optimum signal to noise ratio.

   If the OVERLOAD INPUT lamp flashes, the black INPUT ATTENUATOR knob should be switched anti-clockwise until the overload is removed. The meter deflection should then be restored by turning the transparent knob anti-clockwise.

7. Set the WEIGHTING NETWORK switch of 2209 to "Ext. Filter".

8. Select the desired filter on 1616 by depressing the FILTER SWITCH and turning it to the required position.

9. If the meter deflection is less than 1/3 of full scale, increase sensitivity using the transparent knob.

   If the OVERLOAD OUTPUT lamp flashes, turn the transparent OUTPUT ATTENUATOR knob until the overload is removed. If the overload cannot be removed while retaining a meter deflection, any reading taken will be lower than the true signal level and the overload indication should be noted with the results.

10. The RMS vibration level in the third octave band selected is read on the meter. The level corresponding to full scale deflection is indicated by the number which appears between the two red lines on the transparent knob.

3.5. RMS MEASUREMENTS AND STATISTICAL ACCURACY

In any true RMS measurement system the accuracy of the data obtained is not only dependent on the system accuracy but also on the inaccuracies arising from the effect of statistically random signals on the system's averaging process. To understand how these inaccuracies arise it is necessary to consider the RMS value of a signal which is defined:

$$\psi = \sqrt{\lim_{T \to \infty} \frac{1}{T} \int_0^T x^2(t)\, dt} \qquad 3.1$$

where T is referred to as the averaging time
x(t) is a time varying signal
ψ is the RMS value of x over an averaging time T

It can be seen from equation 3.1 that to obtain the true RMS value of a statistically random signal the averaging time would have to be infinitely long. Of course, in practice this is impossible to realize and consequently the RMS value of a random signal will fluctuate when displayed on a meter or other readout device employing a more practical averaging time. The shorter the averaging time, the greater will be the fluctuations.

Equation 3.1 also shows that the level of the RMS fluctuation depends on the measurement bandwidth, since if the averaging time is kept constant and the measurement bandwidth is decreased, the level of the RMS fluctuation will increase.

A simple relationship providing a measure of the statistical RMS fluctuation (i.e. the possible range of error in the averaged RMS signal) is:

$$\epsilon = \frac{1}{2\sqrt{BT}} \qquad 3.2$$

where B is the measurement (or signal) bandwidth (whichever is the smallest) in Hz, and T is the averaging time or signal duration (whichever is the smallest) in seconds. This expression is an approximate relation but carries sufficient accuracy for BT products greater than 5. It expresses the limits of signal variation to a confidence level of approx. 68%. To obtain higher confidence levels the approximate relations apply that the variation limit of $\pm 2\,\epsilon$ provide a confidence level of 95% and $+ 2.5\,\epsilon$ provides a confidence limit of 99%. For further details on the derivation of equation 3.2 and its application, refer to "Measurement and Analysis of Random Data" by J.S. Bendat and A. G. Piersol (John Wiley 1967) and to statistical textbooks on the chi square distribution.

Equation 3.2 shows that if the averaging time is decreased and the level of RMS fluctuation is increased, the signal can be estimated as random. However, if the level remains roughly the same, the signal can be estimated as periodic. Also, if the measurement bandwidth is decreased to increase the frequency resolution then the averaging time must be increased to obtain the same measurement accuracy.

It can be seen therefore that equation 3.2 provides a useful relationship for determining a practical averaging time for minimizing the fluctuations.

The three meter time constants of the 2209 (two for 2203) provide three (or two) effective averaging times for insertion into equation 3.2. For the purpose of determining effective statistical error in RMS measurements of random noise, the following averaging times may be used.

Fast	0.4 s
Slow	1.0 s
Impulse (rise)	0.07 s

For a 1/3 octave analysis of random noise using 1616 with a 2203 or 2209, the RMS fluctuations will be as shown in Fig.3.6. It will not be possible to reduce these fluctuations using the standard rectifiers of the 2203 or 2209.

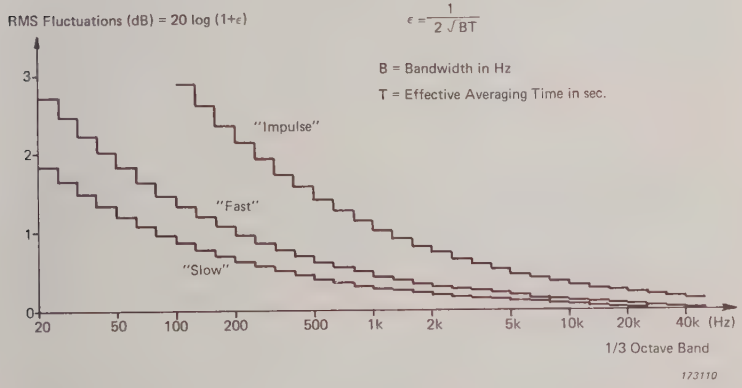

Fig.3.6. RMS fluctuation from true RMS value of a random signal measured in 1/3 octave bands using 1616 and 2203 or 2209. Only "Fast" and "Slow" characteristics apply to 2203

4. DESCRIPTION

4.1. GENERAL

A block diagram of the 1/3 Octave Filter Set Type 1616 is shown in Fig.4.1.

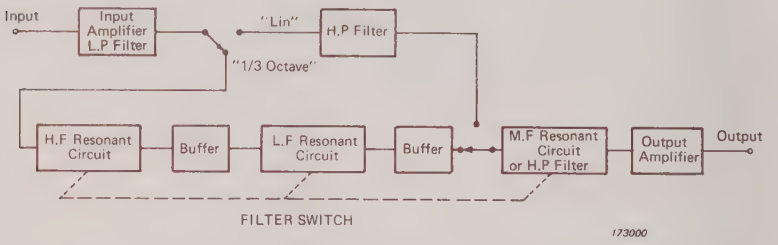

Fig.4.1. Block diagram of 1616

All filters in the 1616 are active filters. The Filter Set includes 34 third-octave effective bandwidth filters with centre frequencies ranging from 20 Hz to 40 kHz. Filter shift is made by changing resistors and capacitors in the selective stages so that the same active elements are used for all positions of the FILTER SWITCH.

A linear range (22.4 Hz — 40 kHz) is also included.

4.2. INPUT

Connection to the IN socket is made via the Connection Bar JP 0400 or the B & K Coaxial Plug JP 0006.

The input stage is a buffer providing high input impedance (10 kΩ) and low driving impedance for the first selective stage. The input stage also acts as an active low-pass filter to damp unwanted high frequency

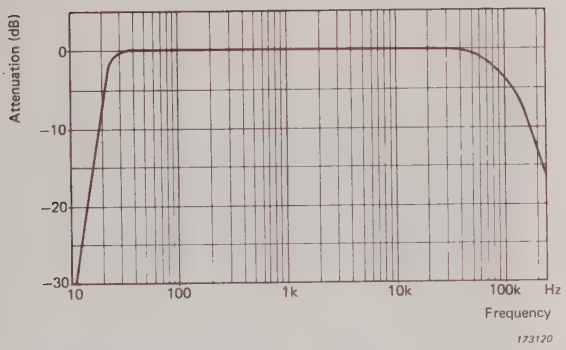

Fig.4.2. Typical frequency response of 1616 in the "Lin" mode

components. It is designed for linear phase response, giving a smooth frequency cut-off and impulse or transient response without overshoot. This last feature is very important when the 1616 is used in its "Lin" mode in conjunction with an impulse sound level meter. The frequency and phase responses of 1616 in the "Lin" mode are given in Fig.4.2 and 4.3 respectively. The high frequency response is determined by the input stage, the low frequency end being determined by a high-pass filter described in § 4.4.

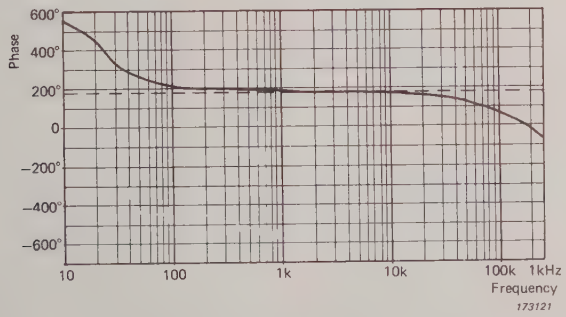

Fig.4.3. Typical phase response of 1616 in the "Lin" mode

4.3. 1/3 OCTAVE FILTERS

4.3.1. General

The 1/3 octave filters are all 6-pole Butterworth filters with centre frequencies and bandwidths given in Table 4.1; complete filter characteristics as regards to band-edge frequencies, intersection frequencies, bandwidth, etc. are given in Table 4.2. The centre frequencies range from 20 Hz to 40 kHz according to the preferred frequencies specified in ISO R 266.

Centre Frequency Hz	Bandwidth Hz	Centre Frequency Hz	Bandwidth Hz	Centre Frequency Hz	Bandwidth Hz
20	4.62	250	58.2	3150	732
25	5.82	315	73.2	4000	922
31.5	7.32	400	92.2	5000	1160
40	9.22	500	116	6300	1460
50	11.6	630	146	8000	1840
63	14.6	800	184	10000	2310
80	18.4	1000	231	12500	2910
100	23.1	1250	291	16000	3670
125	29.1	1600	367	20000	4620
160	36.7	2000	462	25000	5820
200	46.2	2500	582	31500	7320
				40000	9220

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Table 4.1. Third octave centre frequencies and bandwidths

The filters are designed to have an Effective Bandwidth of one third octave (i.e. they pass the same amount of white noise power as an ideal filter of one third octave bandwidth). With the filter shape used, the band-edge frequencies correspond to an attenuation of nominally 3.65 dB. In order to meet the requirements of ANSI S.1.11-1966, a third order filter is used. With 1616, this is made using three stagger tuned resonant circuits. Note that the third stage (mean frequency resonant circuit) is used as a high-pass filter in the "Lin" mode. The typical frequency characteristics of the three resonant circuits and the resulting 1/3 octave filter characteristics are given in Fig.4.4.

		Hz									
Nominal Centre Frequency	f_o	10	12.5	16	20	25	31 5	40	50	68	80
Exact Centre Frequency	$f_m = 10^{n/10}$	10	12.5893	15.8489	19.9526	25 1189	31.6228	39.8107	50 1187	63.0957	79.4328
Upper Band-Edge Frequency	+ 1/6 octave (−3.65 dB)	11.2246	14.131	17.7898	22.3961	28.1950	35.4954	44.6860	56.2564	70.8226	89.1603
Lower Band-Edge Frequency	−1/6 octave (−3.65 dB)	8.90899	11.2158	14.1198	17.7758	22.3784	28.1727	35.4673	44.6507	56.2119	70.7666
Upper Intersection Frequency	+ 1/20 decade (−3.60 dB)	11.2202	14.1254	17.7872	22.3872	28.1838	35.4813	44.6684	56.2341	70.7946	89.1251
Lower Intersection Frequeney	−1/20 decade (−3.60 dB)	8.91251	11.2202	14.1254	17.7828	22.3872	28.1838	35.4813	44.6684	56.2341	70.7946
Bandwidth at Band-Edge Frequencies	Δf 23.1561% of f_m	2.31561	2.9152	3.6700	4.6203	5.8166	7.3227	9.2187	11.6057	14.6107	18.3937
Bandwidth at Intersection frequencies	Δf 23.0769% of f_m	2.30769	2.9052	3.6574	4.6044	5.7966	7.2975	9.1871	11.5657	14.5605	18.3305
Bandwidth at −3 dB points	Δf 22.1201% of f_m	2.21201	2.7847	3.5058	4.4135	5.5563	6.9950	8.8061	11.0863	13.9568	17.5704
Resonant Frequency HF Resonant Circuit	f_r (HF)	11.0052	13.8547	17.4421	21.9583	27.6438	34.8015	43.8125	55.1567	69.4381	87.4174
Resonant Frequency LF Resonant Circuit	f_r (LF)	9.08668	11 4395	14.4014	18 1303	22 8247	28 7346	36 1747	45 5413	57 3331	72 1781
Resonant Frequencies for combined HF and LF circuits (Fig.4.4)	f_r (HF)	10 8125	13 6121	17 1367	21 5738	27 1598	34 1921	43 0453	54 1909	68 2223	85 8867
	f_r (LF)	9.24855	11.6432	14.6580	18.4533	23.2313	29.2465	36.8192	46.3526	58.3544	73.4639
Pole Location	(HF)	− 3.80635 ± j 69.0928	− 4.79192 ± j 86.9197	− 6.03266 ± j 109.425	− 7.59467 ± j 137.758	− 9.56113 ± j 173.428	− 12.0367 ± j 218.332	− 15.1534 ± j 274.864	− 19.0770 ± j 346.034	− 24.0165 ± j 435.63	− 30.2349 ± j 548.426
	(MF)	− 6.9492 ± j 57.0067	− 8.99853 ± j 78.6157	− 11 0137 ± j 98.9713	− 13 8655 ± j 124.598	− 17 4556 ± j 156.859	21 9753 ± j 197.474	27 6653 ± j 248.605	− 34 8285 ± j 312.875	− 43 8965 ± j 394.012	− 55 1995 ± j 496.032
	(LF)	− 3.14285 ± j 57.0067	− 3.95661 ± j 71.7672	− 4.98108 ± j 90.3495	− 6.27081 ± j 113.743	− 7.89442 ± j 143.194	− 9.93856 ± j 180.271	− 12.5119 ± j 226.948	− 15.7516 ± j 285.710	− 19.8300 ± j 359.688	− 24.9645 ± j 452.820

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Table 4.2. Filter parameters for Type 1616

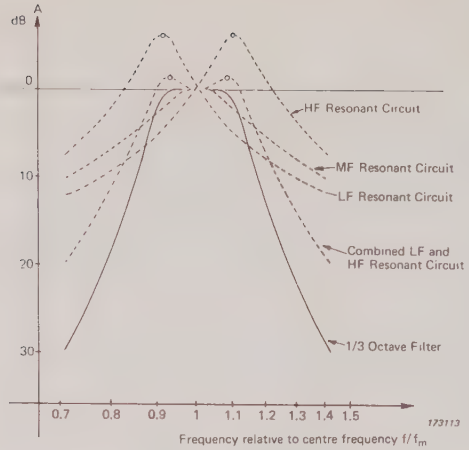

Fig. 4.4. Typical frequency response of the three resonant circuits and the resulting third octave filter characteristics

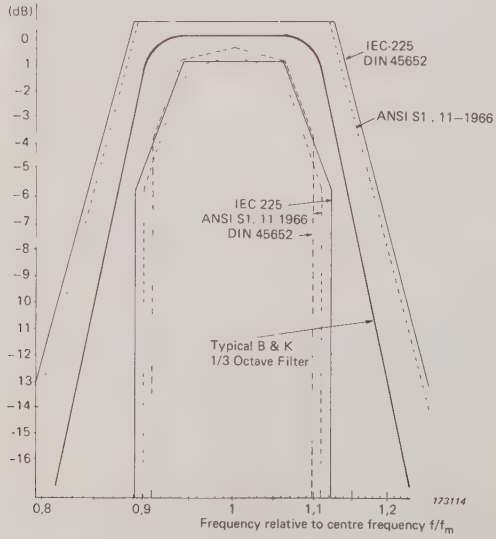

Fig. 4.5. Pass-band of a typical third octave filter relative to IEC, ANSI and DIN recommendations

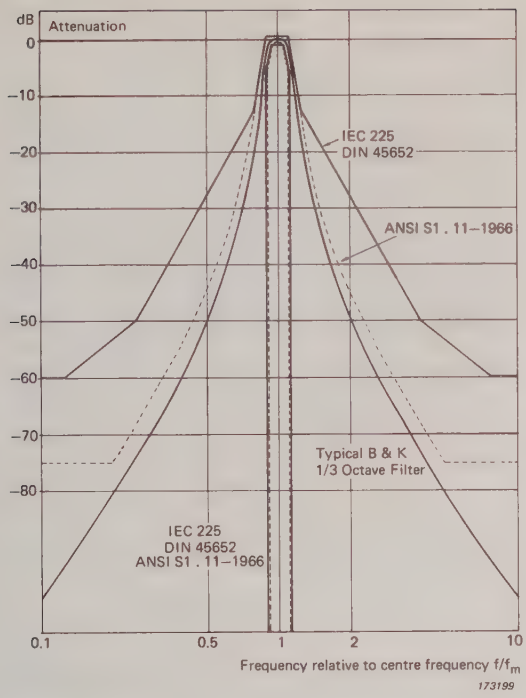

Fig.4.6. Characteristics of a typical third octave filter relative to IEC, ANSI and DIN recommendations

The overall characteristics of the third octave filters conform to the requirements of ANSI S1.11-1966 class III, IEC Recommendation 225 and DIN 45652. The pass band of a typical filter relative to these requirements is shown in Fig.4.5, while the complete characteristics are given in Fig.4.6. Peak-to-valley ripple in the pass-band is less than 0.5 dB, and rejection at frequencies outside one fifth and five times the centre frequency is a greater than 75 dB.

The phase response of the filters is shown in Fig.4.7. The solid line shows the typical response, while the dotted lines indicate the range of likely variations between filters.

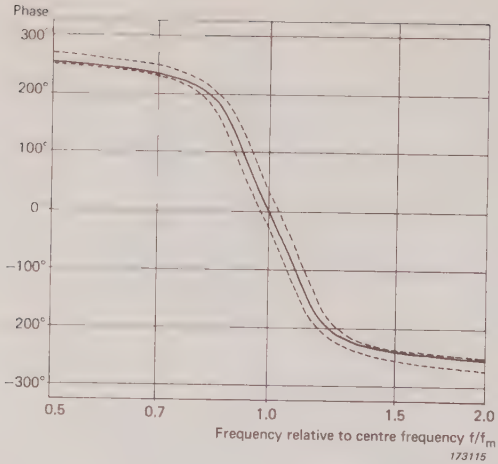

Fig.4.7. Phase response of a typical third octave filter

4.3.2. Filter design

The filter transfer function may be written as:

$$H(p) = K \cdot \frac{p^3}{(p-p_1)\,(p-p_2)\,(p-p_3)\,(p-p_4)\,(p-p_5)\,(p-p_6)}$$

where p_1 to p_6 are the poles of the system. Location of the complex conjugate pole pairs are shown in Fig.4.8 where the filter centre frequency is obtained for $\omega = 1$.

Each pair of complex conjugate poles corresponds to one resonant circuit. The resonant circuit is an active filter built around an NIC (Negative Impedance Converter) element. The basic resonant circuit is shown in Fig.4.9a.

Since the filter shift is made by changing the components marked with an arrow on Fig.4.9a, it is important that the output impedance of the driving circuit is so low that its influence is negligible. Also, because of the high output impedance of the resonant circuit in the pass-band, both the load impedance and input impedance of the operational amplifier must be high. A buffer stage is therefore inserted between the resonant circuits (see block diagram, Fig.4.1) and FET stages are placed

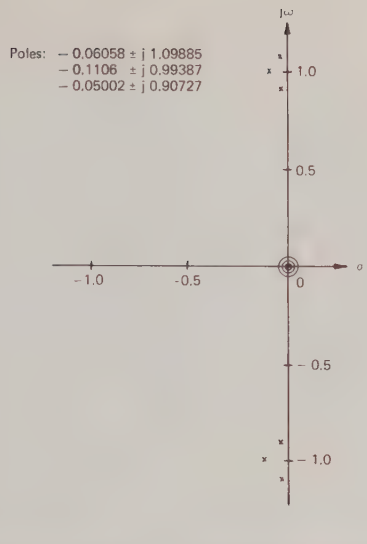

Fig.4.8. Pole and zero location of a third octave filter transfer function

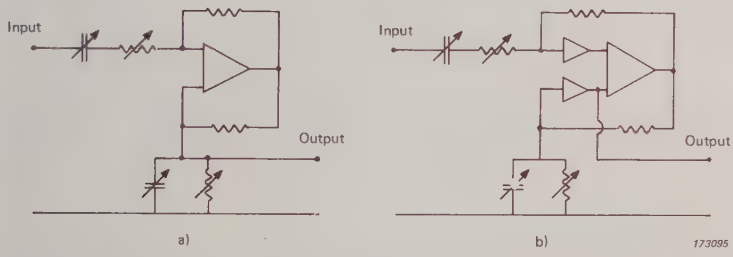

Fig.4.9. a) Basic resonant circuit, b) Practical resonant circuit

at the inputs of the operational amplifier as shown in Fig.4.9b, where the output is now taken at the low impedance output of the FET stage.

The buffer stages also control the voltage level in the filter set. In order to obtain the best possible signal to noise ratio, the buffer stages, as well as the input stage, attenuate the signal which is amplified in the instrument as shown in Fig.4.10.

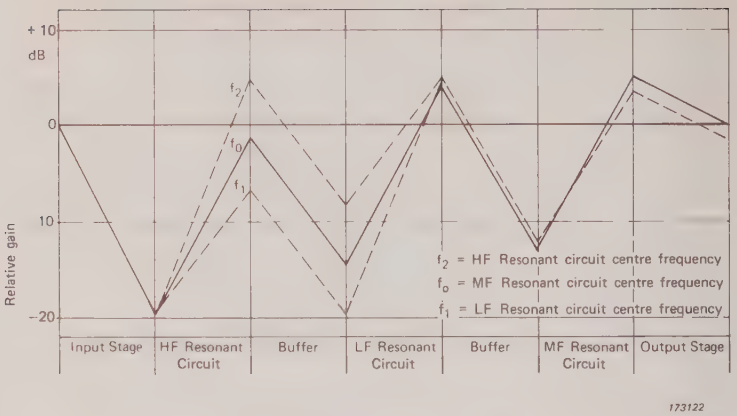

Fig.4.10. Signal level at different stages of 1616

The resonant circuits are designed for equal values of the resistor and capacitor pairs (marked with an arrow in Fig.4.9a and 4.9b). This equality is the critical point in the circuit. The resistors are made by thick-film process. Since they are made from the same paste, on the same substrate, in the same printing and burned together, they will, after adjustment, be very closely equal over a wide temperature range. Careful selection of the capacitors ensures similar features.

Each filter has its own resistor module whereas the capacitors are common for several filters, allowing the rather compact construction of the instrument.

4.3.3. Filter transient response

The transient and step response of the filters are determined by the poles and the zeros of the system. For the chosen Butterworth configuration, the response will be as shown in Figs.4.11 and 4.12. Fig.4.11 shows the impulse response of the filter. Fig.4.12 shows the response to a suddenly applied sinusoidal signal at the filter centre frequency. It can be seen that the output signal will reach its maximum after approx. 6 periods and will have nearly reached the steady state after approx. 15 periods. The maximum level is approx. 8% higher than the steady state level. Fig.4.12 also shows the response when a steady state sinusoidal signal at the filter centre frequency is suddenly suppressed.

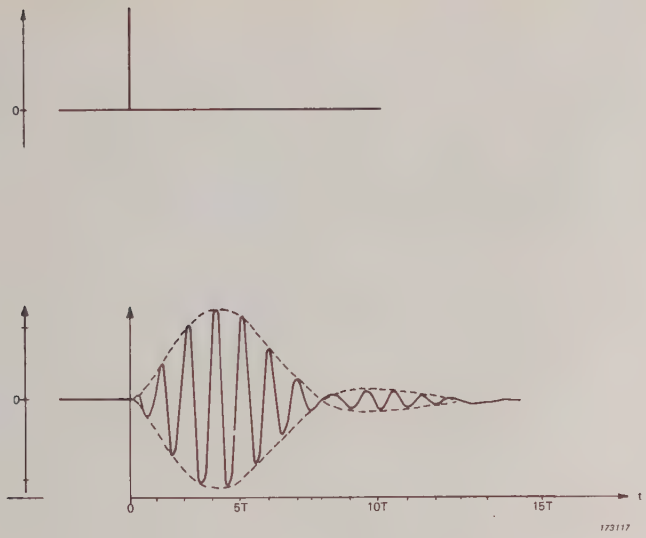

Fig.4.11. Filter impulse response

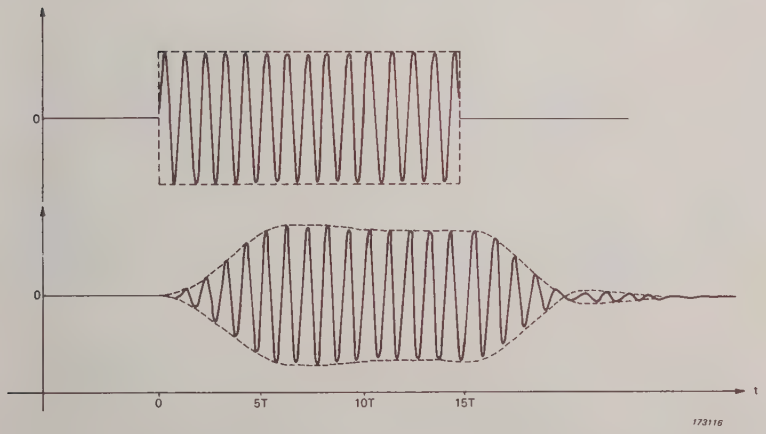

Fig.4.12. Filter transient response to sinusoidal burst at centre frequency

4.4. HIGH-PASS FILTER

When the FILTER SWITCH is in its "Lin" position, the signal from the input stage is fed to the high-pass filter, (see Fig.4.1). It is a 5-pole Butterworth filter with the 3 dB point at 22.4 Hz. Fig.4.13 shows the typical frequency response of the high pass filter and the tolerance limits, while the overall frequency and phase response of the 1616 in the "Lin" mode are given in Figs.4.2 and 4.3. The pole and zero location of the high-pass filter is given in Fig.4.14.

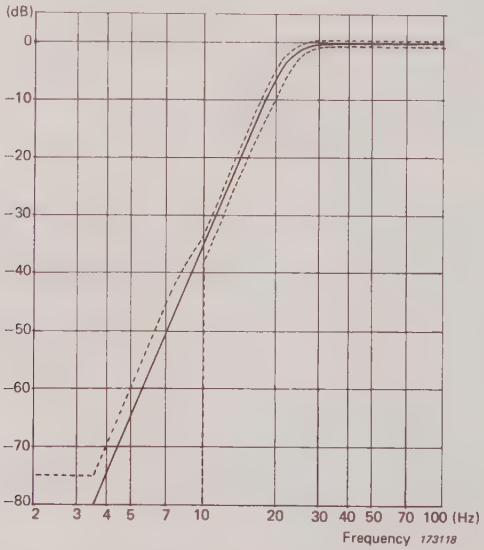

Fig.4.13. Typical frequency response of the high-pass filter and tolerance limits

As can be seen from the block diagram in Fig.4.1, the high-pass filter is divided into two sections. The first section is special for the filter. It is a controlled voltage source active filter giving three (one real and two complex conjugates) of the 5 poles of the filter. The second section uses the last stage of the selective section to generate the last two complex conjugate poles.

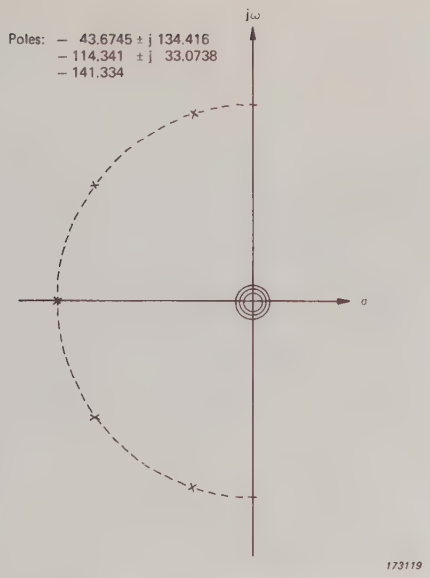

Fig.4.14. Pole and zero location of the high-pass filter transfer function

4.5. OUTPUT AMPLIFIER

The output amplifier acts as a buffer stage between the last resonant circuit and the external load. Output impedance is below 5 Ω in series with 0.47 μF. The minimum load impedance is 2 kΩ.

If the FILTER SWITCH is depressed, an attenuation of approx. 30 dB is applied to the signal so that transients due to filter switching will not overload unnecessarily the amplifiers and detectors connected to the filter output.

4.6. POWER SUPPLY

The 1616 is powered from 3 standard 1.5 V batteries (D cells, type R 20 in IEC Publication 86-2. B & K part number QB 0004). A DC to DC converter provides the necessary symmetrical voltages for the operational amplifiers. The converter is stabilized for battery voltages

between 3 and 4.6 V. The oscillation frequency of the circuit is about 50 kHz.

4.7. NOISE

The inherent noise spectrum of the 1616 is given in Fig.4.15, while the noise spectra of the combination of the Filter Set with the Precision Sound Level Meter Type 2203 and the Impulse Precision Sound Level Meter Type 2209 are given in Fig.4.16 and 4.17 respectively.

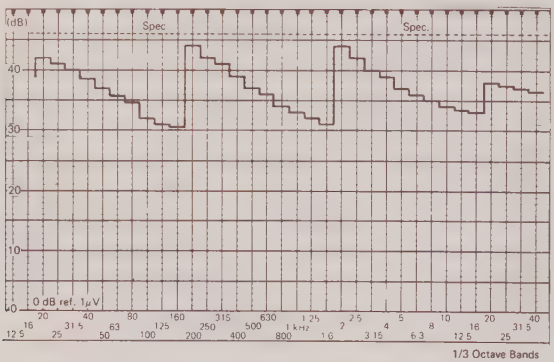

Fig.4.15. Inherent noise of 1616

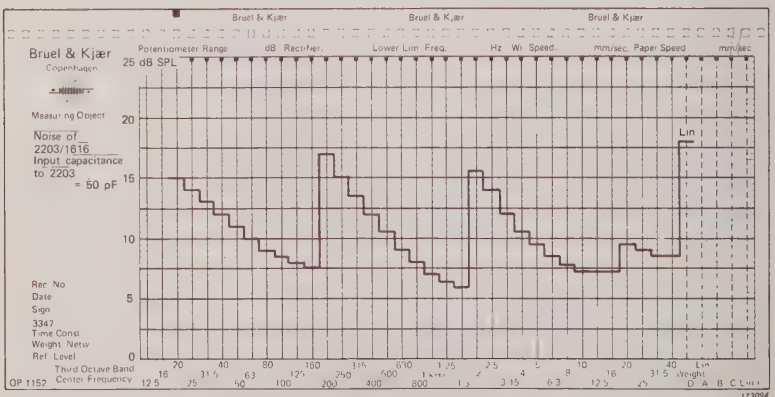

Fig.4.16. Equivalent noise levels referred to input for 1616 and 2203 (with a one inch microphone)

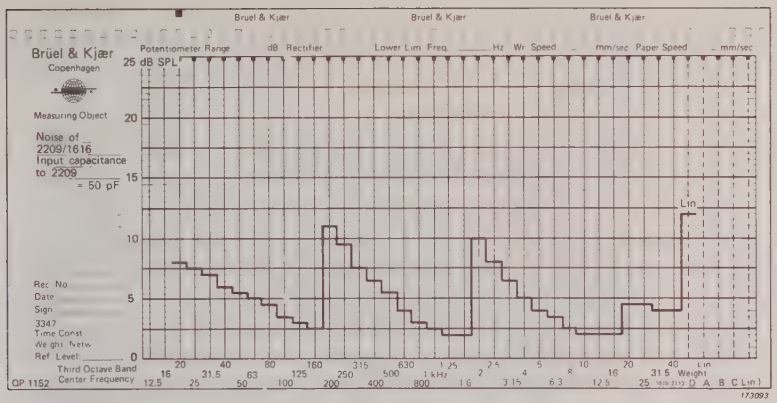

Fig. 4.17. Equivalent noise levels referred to input for 1616 and 2209 (with a one-inch microphone)

5. APPLICATIONS

5.1. GENERAL

The 1/3 Octave Filter Set Type 1616 can be used, in conjunction with the Precision Sound Level Meter Type 2203 or the Impulse Precision Sound Level Meter Type 2209, for a very wide range of applications both for acoustic and vibration measurements and analysis. Only a few typical applications will be described below.

5.2. LOUDNESS EVALUATION

The 1/3 octave spectrum may be used for loudness evaluation according to the procedure suggested by E. Zwicker and described in ISO Recommendation R 532.

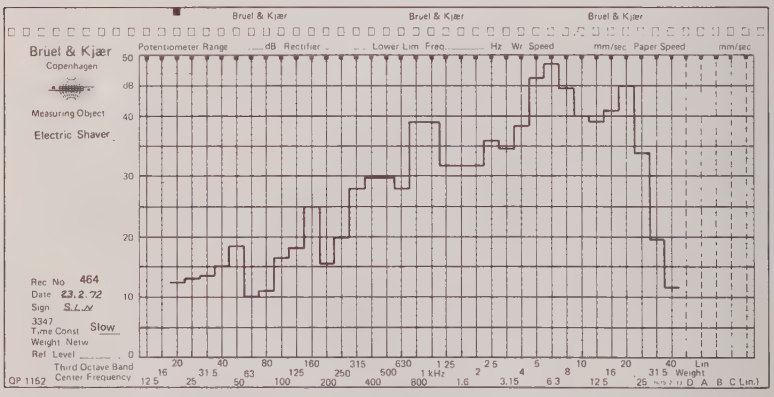

Fig.5.1. 1/3 octave spectrum of the noise from an electric shaver

The 1/3 octave spectrum levels of Fig.5.1 are transformed into a "Zwicker Diagram" (Fig.5.2) corresponding to the measurement

conditions. For the full procedure for loudness evaluation, reference should be made to the ISO document. The level of loudness in phones (GF or GD) is calculated from the total area and is read from the scale to the right of the diagram (Fig.5.2).

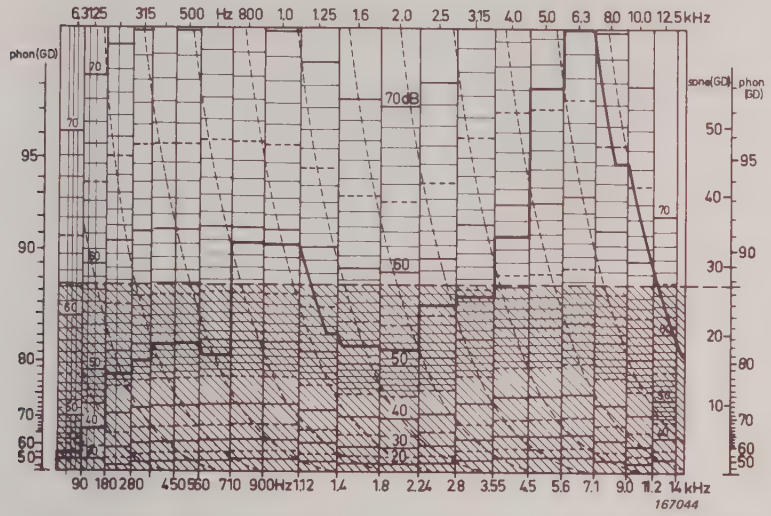

Fig.5.2. The spectrum of Fig.5.1 converted for loudness calculation

5.3. ACOUSTIC IMPULSE LOUDNESS EVALUATION

Calculation of the loudness of impulsive sounds may be carried out using a method described by D.R. Johnson and D.W. Robinson. The 1/3 octave "energy" spectrum (sound pressure squared x time) is converted into an equivalent 1/3 octave band pressure level spectrum, assuming an integration time of the ear of 70 ms. These levels are then converted by means of the chart of Fig.5.3 and the loudness indices are added according to the formula:

$$S_t = S_m + 0.15\,(\Sigma S - S_m)$$

where S_m is the largest of the indices and ΣS the sum of all the indices. Using the scale to the right of Fig.5.3, the total loudness value (sones) can be converted into a total loudness level (phons).

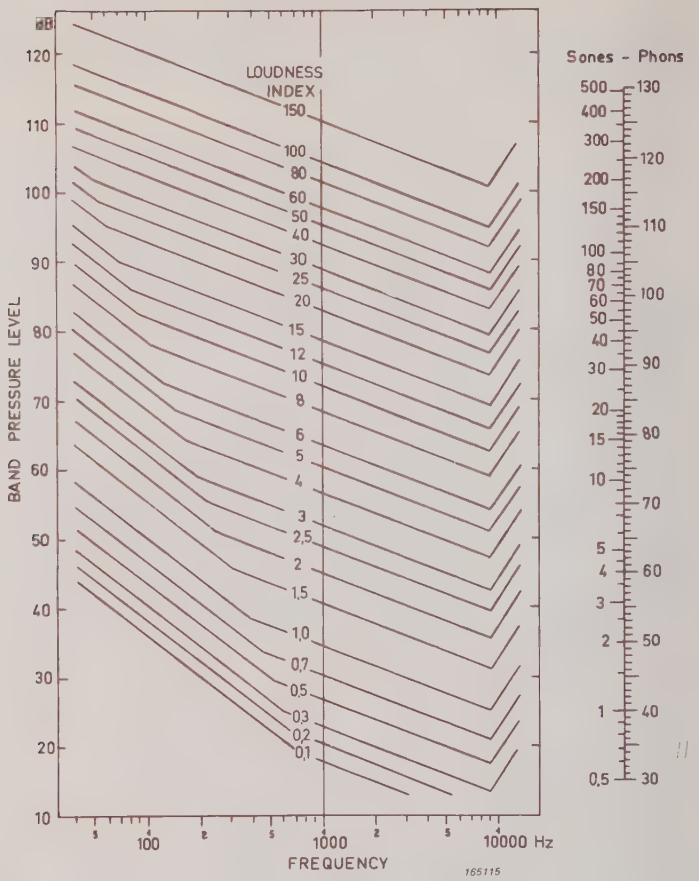

Fig.5.3. Contours of equal loudness index

An alternative procedure is to use an Impulse Precision Sound Level Meter Type 2209 with the "Impulse" or "Impulse Hold" characteristic together with the 1/3 Octave Filter Set Type 1616. This gives directly a band pressure level spectrum and avoids the rather time-consuming conversion of the energy spectrum into the band pressure level spectrum. The calculation is then as in the previous method.

Full description and comparison of the two methods may be found in the B&K Technical Review No.1-1972, "Loudness Evaluation of Acoustic Impulses" by P. Hedegaard.

5.4. REVERBERATION TIME MEASUREMENTS

The Reverberation Processor Type 4422 allows reverberation measurements according to the Schroeder-Kuttruff method of impulsive excitation, giving very smooth decay curves. Type 4422 also provides direct reading of the EDT (Early Decay Time) which is of special importance when evaluating the acoustical quality of an enclosure.

Due to its low output impedance, the 1/3 Octave Filter Set Type 1616 may be used for such measurements in conjunction with a Power Amplifier Type 2706. An example of set-up is given in Fig.5.4. For further details on the Reverberation Processor and the Schroeder-Kuttruff method, reference whould be made to the instruction manual for Type 4422.

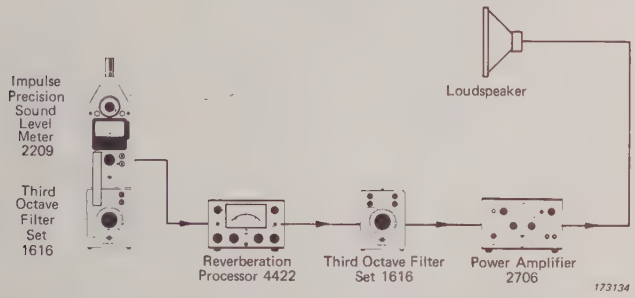

Fig.5.4. Set-up for measuring EDT using 1616 in conjunction with the Reverberation Processor Type 4422

Type 1616 may also be used for reverberation measurements according to the interrupted noise method, as well as any building acoustics measurements where a 1/3 Octave Filter Set is needed.

5.5. MEASUREMENT OF HARMONICS IN EARPHONES AND HEARING AIDS

Measurement of harmonic distortion in earphones and hearing aids may be carried out using the 1616 in conjunction with a Sound Level Meter.

For measurement of harmonic distortion in hearing aids, use can be made of the Hearing Aid Test Box Type 4217 which has 15 fixed frequencies with one third octave spacing from 200 Hz to 5 kHz. The use of 1616 and 4217 for harmonic distortion measurements is described in the instruction manual for the Hearing Aid Test Box.

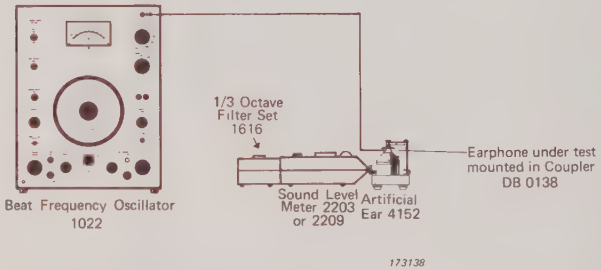

Fig.5.5. Set-up for measuring frequency characteristics of fundamental and harmonic signals in earphones

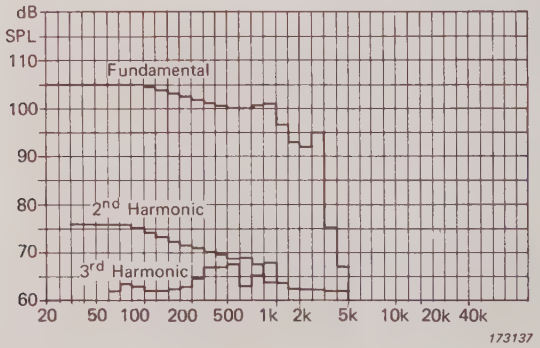

Fig.5.6. Typical frequency characteristics of fundamental and harmonic signals in an earphone

For measurement of harmonic distortion in earphones, use can be made of the set-up shown in Fig.5.5. The generator is a low distortion Beat Frequency Oscillator Type 1022. The Artificial Ear Type 4152 is used as the receiver in connection with a Sound Level Meter Type 2203 or 2209. An example of harmonic distortion measurements is given in Fig.5.6.

6. ASSOCIATED INSTRUMENTATION AND ACCESSORIES

6.1. PRECISION SOUND LEVEL METER TYPE 2203

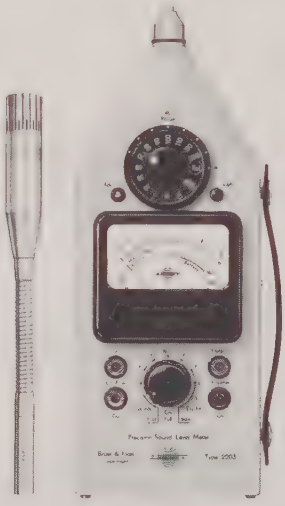

Fig.6.1. The Precision Sound Level Meter Type 2203

The Precision Sound Level Meter Type 2203 is a compact, portable, battery operated instrument which fulfils the requirements of IEC Recommendation 179 and DIN 45633 part 1 for precision sound level meters.

Type 2203 has a frequency range (± 2 dB) of 10 Hz to 18 kHz with a one-inch Condenser Microphone Type 4145 and of 10 Hz to 25 kHz with a half-inch Condenser Microphone Type 4133. A, B and C weighting networks are incorporated as well as the normalized "Fast"

and "Slow" meter dampings. The quasi-RMS detector of the 2203 has a crest factor capability of 3.

The 1/3 Octave Filter Set Type 1616 screws directly on the 2203, making a portable sound and vibration analyzer. Note that for a 2203 with a serial No. 424272 and earlier, a special bottom plate, part No. DD 0135 must be ordered separately.

Further details on the 2203 may be found in its own instruction manual.

6.2. IMPULSE PRECISION SOUND LEVEL METER TYPE 2209

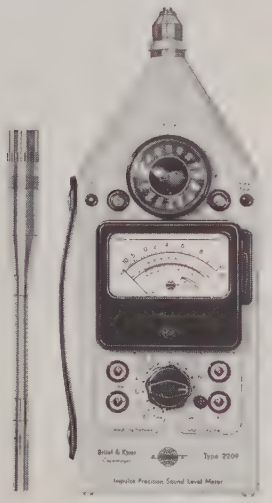

Fig.6.2. The Impulse Precision Sound Level Meter Type 2209

The Impulse Precision Sound Level Meter Type 2209 has the same external dimensions as Type 2203. It is a compact, portable, battery operated instrument which conforms to IEC 179 for precision sound level meters, the proposed IEC Recommendation for impulse sound level meters, and to DIN 45633, parts 1 and 2.

Type 2209 has a frequency range of 3 Hz to 18 kHz ($\pm^{1}_{1.5}$ dB) with a one inch Condenser Microphone Type 4145 and of 4 Hz to 40 kHz (± 2 dB) with a half-inch Condenser Microphone Type 4133. A, B, C and D weighting networks are incorporated. The 2209 has the "Fast" and "Slow" meter dampings as well as an "Impulse" response with an integration time constant of 35 ms. It also includes a hold circuit which will retain the maximum RMS level of the input signal in the "Imp. Hold" mode or the maximum peak level of the input signal in the "Peak Hold" mode. Crest factor capability is up to 40. 10 double sided interchangeable attenuator scales supplied with the instrument allow direct reading of sound level or vibration units with different transducers.

The 1/3 Octave Filter Set Type 1616 screws directly on Type 2209, making a portable noise and vibration analyzer. The noise spectrum of the combined instruments may be found in Fig.4.17.

Further details on the Impulse Precision Sound Level Meter may be found in the instruction manual for Type 2209.

6.3. ACCESSORIES

6.3.1. Connection Bar JP 0400

Fig.6.3. Connection Bar JP 0400

The Connection Bar JP 0400 is used to electrically connect the 1616 and the Sound Level Meter. It is supplied as standard accessory with the Filter Set.

6.3.2. Plug JP 0006

Fig.6.4. JP 0006

This coaxial plug fits the IN and OUT sockets of the 1616 and the EXT. FILTER sockets of the Sound Level Meters. Two of them are supplied with the 1616.

6.3.3. Connection Cable AO 0007

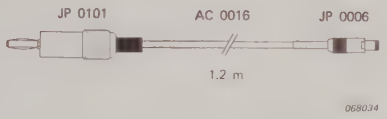

Fig.6.5. Connection Cable AO 0007

This cable connects the 1616 to other B & K instruments with standard B & K coaxial sockets.

6.3.4. Bottom Plate DD 0135

The Bottom Plate DD 0135 should be ordered separately if the 1616 is to be used with a Precision Sound Level Meter Type 2203 with serial No. 424272 and earlier. Note that the bottom plate should be mounted with the tapped hole to the upper edge.

7. USEFUL FILTER DEFINITIONS

7.1. FILTER DEFINITIONS

The values of the quantities defined below may be found, for the 1616, in Table 4.2.

7.1.1. Ideal Filter (Band Pass)

An ideal band pass filter is one which has zero attenuation within its pass-band and infinite attenuation at all other frequencies. The ideal filter compared with a practical filter in Fig.7.1.

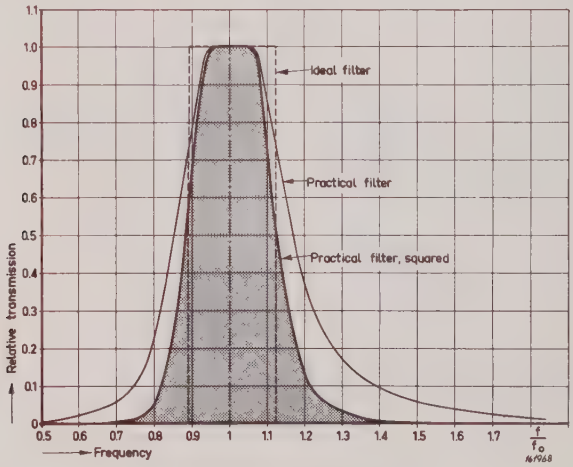

Fig.7.1. Ideal and Practical Filters

7.1.2. Octave

An octave is the interval between two pure tones having a frequency ratio of two, i.e.

$$f_U = 2 f_L \text{ or } \frac{f_U}{f_L} = 2$$

Fractional octaves are similarly defined by $f_U / f_L = 2^n$, where n is the fraction of the octave. Third octaves are therefore defined by $f_U / f_L = 2^{1/3} = 1.259$. Table 7.1 gives the frequency ratios for different values of n.

7.1.3. Band-Edge Frequencies

The band-edge frequencies are the frequencies at which the filter response becomes a certain value (often chosen as 3 dB) below the maximum transmission level.

	OCTAVE	Frequency ratio	
		up	down
1/12	0.0833	1.0594	0.0439
1/8	0.1250	1.0905	0.9170
1/6	0.1667	1.1225	0.8909
1/4	0.2500	1.1892	0.8409
1/3	0.3333	1.2591	0.7937
1/2	0.5000	1.4142	0.7071
3/4	0.7500	1.6817	0.5946
1	1.0000	2.0000	0.5000
	n	2^n	2^{-n}

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Table 7.1. Band-edge frequency ratios for fractional octaves

7.1.4. Centre (or Mean) Frequency

The centre frequency of a pass-band is the geometric mean of the band-edge frequencies, i.e.

$$f_m = \sqrt{f_U f_L} \quad \text{(Hz)}$$

7.1.5. Bandwidth

The bandwidth of a pass-band filter is the difference between the upper and lower band-edge frequencies, i.e.

$$B = f_U - f_L \quad \text{(Hz)}$$

The bandwidth may be expressed in octaves, i.e.

$$b_1 = \log_2 \frac{f_U}{f_L} \text{ (octaves)}$$

or as a percentage of the pass-band centre frequency

$$b_2 = \frac{f_U - f_L}{f_m} \cdot 100 \ (\%)$$

7.1.6. Effective Bandwidth

The effective bandwidth of a filter is equal to the bandwidth of an ideal filter that would pass the same amount of white noise power as the practical filter.

7.1.7. Spectrum

The spectrum of a function of time is a description of its resolution into components, each of different frequency.

Continuous Spectrum — A continuous spectrum is a spectrum in which the components are continuously distributed over a frequency range.

White Noise Spectrum — A white noise spectrum is a continuous spectrum with a constant spectrum density (mean square amplitude per unit frequency) over a specified frequency range.

Line Spectrum — A line spectrum is a spectrum consisting of discrete single frequency components. components.

7.2. APPLICATION TO 1616

The filters in Type 1616 have an effective bandwidth of one third octave. This bandwidth is given as well as the corresponding 3 dB bandwidth and the bandwidth defined by the intersection frequencies, in Table 4.2.

8. SPECIFICATIONS

Frequency Range:	Selective: 18 Hz — 44 kHz Linear: 22.4 Hz — 40 kHz
1/3 Octave Filters:	
Centre frequencies:	20 Hz — 40 kHz, 34 filters
Attenuation:	0 dB ± 0.5 dB at centre frequency > 75 dB at 5 x and 1/5 x centre frequency
Peak-to-valley ripple:	< 0.5 dB

Each filter satisfies the requirements of IEC Recommendation 225, ANSI S1.11-1966 1/3 Octave Band Filters class III and DIN 45652.

Linear Range:	
Attenuation:	0 dB ± 0.5 dB
Frequency response:	see Fig.4.2
Input:	
Input impedance:	10 kΩ (± 5%)
Input voltage:	2 V peak max.
Max. DC voltage on input:	10 V
Output:	
Output impedance:	< 5 Ω in series with 0.47 μF
Min. load impedance:	2 kΩ
Inherent Noise:	< 200 μV RMS (see also Fig.4.15)
Magnetic Sensitivity:	approx. 1 mV at output for 100 A/m field strength
Operating Temperature:	—20°C to + 50°C (—4°F to 122°F)
Effect of Temperature:	< 0.5 dB
Effect of Humidity:	< 0.5 dB (0 to 90% RH)

Batteries	3 x 1.5 V standard type D cells (R 20 or IEC publication 86-2
Battery Life	5 hours for normal operation
Dimensions	Length: 16.5 cm (6.5") Width: 12 cm (4.75") Height: 9 cm (3.5")
Weight	1.9 kg (4.2 lb) including batteries
Accessories Supplied	1 Connection Bar JP 04001 2 Plugs JP 00004 3 Batteries DB 00004

BRÜEL & KJÆR instruments cover the whole field of sound and vibration measurements. The main groups are:

ACOUSTICAL MEASUREMENTS
Condenser Microphones
Piezoelectric Microphones
Microphone Preamplifiers
Sound Level Meters
Precision Sound Level Meters
Impulse Sound Level Meters
Standing Wave Apparatus
Noise Limit Indicators
Microphone Calibrators

ACOUSTICAL RESPONSE TESTING
Beat Frequency Oscillators
Random Noise Generators
Sine-Random Generators
Artificial Voices
Artificial Ears
Artificial Mastoids
Hearing Aid Test Boxes
Audiometer Calibrators
Telephone Measuring Equipment
Audio Reproduction Test Equipment
Tapping Machines
Turntables

VIBRATION MEASUREMENTS
Accelerometers
Force Transducers
Impedance Heads
Accelerometer Preamplifiers
Vibration Meters
Accelerometer Calibrators
Magnetic Transducers
Capacitive Transducers
Complex Modulus Apparatus

VIBRATION TESTING
Exciter Controls – Sine
Exciter Controls – Sine – Random
Exciter Equalizers, Random or Shock
Exciters
Power Amplifiers
Programmer Units
Stroboscopes

STRAIN MEASUREMENTS
Strain Gauge Apparatus
Multi-point Panels
Automatic Selectors

MEASUREMENT AND ANALYSIS
Voltmeters and Ohmmeters
Deviation Bridges
Measuring Amplifiers
Band-Pass Filter Sets
Frequency Analyzers
Real Time Analyzers
Heterodyne Filters and Analyzers
Psophometer Filters
Statistical Distribution Analyzers

RECORDING
Level Recorders
Frequency Response Tracers
Tape Recorders

DIGITAL EQUIPMENT
Digital Encoder
Digital Clock
Computers
Tape Punchers
Tape Readers

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Orboe Print · Copenhagen · Denmark